CLAIMS

What is claimed is:

1. A method for processing audio signals generated by an array of two or more microphones, comprising the steps of:

(a) filtering the audio signal from each microphone to generate a processed audio signal for each microphone and combining the processed audio signals to form an acoustic beam that focuses the array on one or more regions in space; and

(b) performing nonlinear signal estimation processing on the processed audio signals from the microphones to generate an output signal for the array, wherein the nonlinear signal estimation processing discriminates against noise originating at an unknown location outside of the one or more desired regions.

2. The invention of claim 1, wherein step (a) comprises the step of delaying and scaling the audio signal from each microphone.

3. The invention of claim 1, wherein step (a) comprises the step of applying a digital filter corresponding to the inverse of each transfer function from a desired focal point to each microphone to compensate for reverberation in a volume containing the array.

4. The invention of claim 1, wherein the output signal is processed in a feedback loop to generate control signals that adjust the nonlinear signal estimation processing of step (b).

5. The invention of claim 4, wherein the control signals adjust weights applied to the processed audio signals during the nonlinear signal estimation processing of step (b).

6. The invention of claim 5, wherein a weight for each processed audio signal is based on a ratio of power in a speech band to power outside the speech band for the processed audio signal.

7. The invention of claim 4, wherein the output signal is processed in another feedback loop to generate other control signals that adjust the filtering of step (a) to attempt to match each of the processed audio signals.

8. The invention of claim 1, wherein the output signal is processed in a feedback loop to generate control signals that adjust the filtering of step (a).

9. The invention of claim 1, wherein the filtering of step (a) is dynamically adjusted to attempt to match each of the processed audio signals.

10. The invention of claim 9, wherein the filtering of step (a) is dynamically adjusted to attempt to match each of the processed audio signals in amplitude and phase to each other and to the output signal.

11. The invention of claim 1, wherein the nonlinear signal estimation processing picks a representative, central value from the processed audio signals, by altering at least one extreme value from at least one of the processed audio signals.

12. The invention of claim 11, wherein the nonlinear signal estimation processing comprises the step of selecting the representative, central value as a median of the processed audio signals.

13. The invention of claim 11, wherein the nonlinear signal estimation processing comprises the steps of:

(1) adjusting the magnitude of one or more of at least one of the highest and lowest values of the processed audio signals to generate a set of adjusted audio signals; and

(2) selecting the representative, central value as a median or average of the adjusted audio signals.

14. The invention of claim 13, wherein:

step (1) comprises the steps of:

(i) adjusting the value of the n highest values down to match the $(n+1)^{th}$ highest data value, where n is a non-negative integer; and

(ii) adjusting the value of the m lowest values up to match the $(m+1)^{th}$ lowest data value, where m is a non-negative integer; and

step (2) comprises the step of selecting the representative, central value as an average of the processed audio signals.

15. The invention of claim 14, wherein the average is a weighted average.

16. The invention of claim 11, wherein the nonlinear signal estimation processing comprises the steps of:

(1) dropping one or more of the highest and lowest values of the processed audio signals to generate a set of adjusted audio signals; and

(2) selecting the representative, central value as an average of the adjusted audio signals.

17. The invention of claim 16, wherein the average is a weighted average.

18. The invention of claim 1, wherein the nonlinear signal estimation processing treats each set of input values for the processed audio signals independently.

19. The invention of claim 1, wherein the nonlinear signal estimation processing is based on multiple values from each processed audio signal over a period of time.

20. The invention of claim 19, wherein the nonlinear signal estimation processing comprises the step of applying temporal filtering to the input values of each processed audio signal.

21. The invention of claim 20, wherein the nonlinear signal estimation processing further comprises the steps of generating a distance measure between pairs of audio signals and generating the output signal from the one or more audio signals having the smallest distance measures with other audio signals.

22. A machine-readable medium, having encoded thereon program code, wherein, when the program code is executed by a machine, the machine implements a method for processing audio signals generated by an array of two or more microphones, comprising the steps of:

(a) filtering the audio signal from each microphone to generate a processed audio signal for each microphone and combining the processed audio signals to form an acoustic beam that focuses the array on one or more regions in space; and

(b) performing nonlinear signal estimation processing on the processed audio signals from the microphones to generate an output signal for the array, wherein the nonlinear signal estimation processing discriminates against noise originating at an unknown location outside of the one or more desired regions.